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Via: First Class Mail	**Atty Dkt No.:** KHEN-P01-001
Inventor: Lavoie et al.	
Application No.: 09/839485	**Filing Date:** April 20, 2001
Title: AUTO-CALIBRATING SURROUND SOUND SYSTEM	

Documents Filed:

Supplemental Amendment (9 pages)

Amendment Transmittal PTO/SB/21 (1 page)

Amendment Transmittal Letter (1 page)

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I hereby certify that this correspondence is being deposited with the U.S. Postal Service with sufficient postage as First Class Mail, in an envelope addressed to: MS Amendment, Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450, on the date shown below.
Dated: 5-25-2005 Signature: Judith Herrick
(Judith Herrick)

Docket No.: KHEN-P01-001
(PATENT)

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Patent Application of:
Lavoie et al.

Application No.: 09/839485

Confirmation No.: 2615

Filed: April 20, 2001

Art Unit: 2644

For: AUTO-CALIBRATING SURROUND SOUND SYSTEM

Examiner: J. I. Michalski

SUPPLEMENTAL AMENDMENT

MS Amendment
Attn: Examiner Justin Michalski
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

INTRODUCTORY COMMENTS

Supplemental to the amendment filed on September 30, 2004, please amend the above-identified U.S. patent application as follows:

Amendments to the Specification begin on page 2 of this paper.

Amendments to the Claims are reflected in the listing of claims which begins on page 3 of this paper.

Remarks/Arguments begin on page 8 of this paper.

AMENDMENTS TO THE SPECIFICATION

Replace the paragraph between page 22, line 17, and page 23, line 12, with the following rewritten paragraph:

-- Referring now to Fig. 12, before running the four-channel surround sound test, the impulse response for each of the satellite speakers in an open laboratory space is deconvolved using the MLS technique. The system is set up so that the four frequency responses can be compared. However, these measurements are not directly compared to those that are taken in the listening environment, since the microphone placement, sound pressure level at the microphone, and the surrounding acoustic impedances can all be different. Because all four responses are similar, they are plotted in an overlay fashion. Fig 12(a) shows the impulse response of an exemplary satellite speaker (in this case, the front-right speaker in the listening environment). Fig. 12(b) shows [[, as well as the]] four overlaid frequency response magnitudes from four speakers. The time of flight (TOF) delay of approximately 2.2 ms indicates that the distance between the microphone and the speaker in this test was approximately 70 cm. Verifying distances like speaker placement using the experimentally ~~exponentially~~ determined time of flight is a good way to determine if the periodic cross-correlation is extracting the correct time base. The response feature arriving with a delay of approximately 4.3 ms indicates a first reflected signal. Only a selective region of the impulse response is modeled. Selecting the region after the TOF and before the first reflection will isolate the portion of the response known as the anechoic response, which is the direct path between the monitor and the microphone. The sharp drop in frequency response at about 3 kHz will be the most difficult portion of the spectral response to whiten. --

Insert the following paragraph on page 24, after line 9:

-- The overlay of Fig. 14(a) shows the closeness of the simulated and the actual whitened results, in particular for the filter order M = 5. This observation combined with test chamber experiments demonstrate that identifying the system through correlation techniques, creating a matched filter of the "Moving Average" (MA) type, and performing real-time whitening may be implemented in practice. --

AMENDMENTS TO THE CLAIMS

1. (Previously presented) A method of auto-calibrating a surround sound system, comprising the acts of:

producing an electric calibration signal, said calibration signal being a temporal maximum length sequence (MLS) signal,

supplying said calibration signal to an electro-acoustic converter for converting the calibration signal to an acoustic response,

transmitting the acoustic response as a sound wave in a listening environment to an acousto-electric converter for converting the acoustic response received by the acousto-electric converter to an electric response signal,

detecting in the electric response signal a reflected signal and isolating a portion of the response signal between a time of flight signal and the reflected signal,

correlating the isolated portion of the electric response signal with the electric calibration signal to compute filter coefficients, and

processing the filter coefficients together with a predetermined channel response of the electro-acoustic converter to produce a substantially whitened system response.

2. (Original) The method of claim 1, wherein the acoustic response is radiated in the listening environment for a time less than approximately 3 seconds.

3. (Original) The method of claim 1, wherein the surround sound system includes a plurality of audio channels, with each channel having at least one electro-acoustic converter, wherein the substantially whitened response is produced independently for each audio channel.

4. (Currently amended) A method of ~~producing~~ optimizing a matched filter for whitening an audio channel in a listening environment, comprising:

a. producing in the audio channel a test output sound corresponding to a temporal maximum length sequence (MLS) signal,

b. receiving the test output sound at a predetermined location in the listening environment, thereby producing an impulse response,

c. analyzing a correlation between the impulse response and the MLS signal,

d. generating filter coefficients of the matched filter,

e. repeating steps (a) through (d) with at least one other MLS signal having a different temporal maximum length, and

f. optimizing the matched filter by selecting those generated filter coefficients that minimize an error term between a desired filter response of the matched filter producing the whitened audio channel and the filter response produced with the generated filter coefficients when driven by the corresponding ~~adjusting the temporal~~ maximum length ~~of the~~ MLS signal ~~based on the analyzed correlation, and~~

~~generating, from the analyzed correlation, filter coefficients of the matched filter that produce the whitened audio channel~~.

5. (Currently amended) The method of claim 4, wherein ~~analyzing~~ the ~~correlation includes producing~~ filter coefficients represent coefficients of a polynomial model of the impulse response.

6. (Currently amended) The method of claim 4, wherein the filter coefficients are generated by ~~analyzing the correlation includes using~~ an auto regressive (AR) model.

7. (Original) The method of claim 5, wherein generating the filter coefficients includes optimizing a closeness of fit between the polynomial model and the matched filter.

8. (Canceled)

9. (Original) The method of claim 5, further comprising cascading the matched filter with a useful audio signal so as to produce the substantially whitened audio channel.

10. (Currently amended) An auto-calibrating surround sound (ACSS) system, comprising:

an electro-acoustic converter disposed in an audio channel and adapted to emit a sound signal in response to an electric input signal,

a processor generating a test signal represented by a temporal maximum length sequence (MLS) and at least one other test signal represented by a different temporal maximum length sequence, and the processor supplying the test ~~signal~~ signals as [[the]] electric input signals ~~signal~~ to the electro-acoustic converter, [[and]]

an acousto-electric converter receiving the sound signal in a listening environment and supplying [[a]] received electric ~~signal~~ signals to the processor, and

a coefficient extractor which generates filter coefficients of a corrective filter,

wherein the processor correlates the received electric ~~signal~~ signals with the test ~~signal~~ signals and optimizes the corrective filter by selecting those generated filter coefficients that minimize an error term between a desired filter response of the corrective filter that produces a whitened audio response of the audio channel in the listening environment and the filter response produced with the generated filter coefficients, when driven by the corresponding ~~adjusts the temporal maximum length of the~~ MLS signal ~~based on the correlation to determine a substantially whitened response of the audio channel in the listening environment~~.

11. (Currently amended) The ACSS system of claim 10, wherein the processor includes an impulse modeler that produces a polynomial least-mean-square (LMS) error fit between a desired whitened response and the ~~substantially whitened~~ filter response produced with the generated filter coefficients ~~determined from the correlated signals~~.

12. (Canceled)

13. (Currently amended) The ACSS system of claim [[12]] 10, wherein the corrective filter is located in an audio signal path between an audio signal line input and the electro-acoustic converter and cascaded with the audio signal line input.

14. (Currently amended) The ACSS system of claim [[12]] 10, wherein the corrective filter forms a part of the processor.

15. (Previously presented) The ACSS system of claim 10, wherein the processor is a digital signal processor (DSP).

16. (Original) The ACSS system of claim 15, further including an analog-to-digital (A/D) converter that converts an analog audio line input and the electric signal supplied by the acousto-electric converter into temporal digital signals.

17. (Original) The ACSS system of claim 15, further including a digital-to-analog (D/A) converter that converts digital output signals from the DSP to an analog audio line output for driving the electro-acoustic converter.

18. (Canceled)

19. (Previously presented) An auto-calibrating surround sound (ACSS) system, comprising:

an electro-acoustic converter disposed in an audio channel and adapted to emit a sound signal in response to an electric input signal,

a processor generating a test signal represented by a temporal maximum length sequence (MLS) and supplying the test signal as the electric input signal to the electro-acoustic converter, and

an acousto-electric converter receiving the sound signal in a listening environment and supplying a received electric signal to the processor,

wherein the processor detects in the received electric signal a reflected signal and correlates a portion of the response signal between a time of flight signal and the reflected signal with the test signal to compute filter coefficients, said processor processing the filter coefficients together with a predetermined channel response of the electro-acoustic converter to produce a substantially whitened system response.

20. (Previously presented) The ACSS system of claim 19, wherein the processor includes an impulse modeler that produces a polynomial least-mean-square (LMS) error fit between a desired whitened response and the substantially whitened response determined from the correlated signals.

21. (Previously presented) The ACSS system of claim 19, further comprising a coefficient extractor which generates filter coefficients of a corrective filter to produce the substantially whitened response of the audio channel.

22. (Previously presented) The ACSS system of claim 21, wherein the corrective filter is located in an audio signal path between an audio signal line input and the electro-acoustic converter and cascaded with the audio signal line input.

23. (Previously presented) The ACSS system of claim 21, wherein the corrective filter forms a part of the processor.

24. (Previously presented) The ACSS system of claim 19, wherein the processor is a digital signal processor (DSP).

25. (Previously presented) he ACSS system of claim 24, further including an analog-to-digital (A/D) converter that converts an analog audio line input and the electric signal supplied by the acousto-electric converter into temporal digital signals.

26. (Previously presented) The ACSS system of claim 24, further including a digital-to-analog (D/A) converter that converts digital output signals from the DSP to an analog audio line output for driving the electro-acoustic converter.

REMARKS

This amendment is filed after a telephone interview with Examiner Michalski on May 17, 2005, during which Examiner Michalski requested additional support in the specification for the amendments to claims 1, 4, and 10.

The feature "isolating a portion of the response signal" recited in amended claim 1 is supported in the provisional application serial number 60/198,927, which is incorporated in this application by reference in its entirety. The paragraph between page 22, line 17, and page 23, line 12, has been amended to incorporate the supporting language from (numbered) page 101, lines 2-4, of the provisional application. A reference to Fig. 12(b) has also been included, and other informalities have been corrected.

Claims 4 and 10 have been amended to more clearly state that the matched filter and the filter coefficients are optimized by comparing a desired filter response for whitening the audio channels(s) with a filter response produced by filter coefficients generated for different maximum length MLS signals. The subject matter of claim 12 has been incorporated in claim 10, and claim 12 has been canceled. Other dependent claims have been amended to provide a proper reference to those claims from which they depend, and to also provide proper antecedent basis where needed.

Support for the amendments to claims 4 and 10 can be found on page 25, lines 2-3 of the instant specification, e.g. minimizing the summed square error terms (LMS) to generate the coefficients for the matched filter, and in the paragraph added above to the specification, which corresponds substantially to the language found in Paragraph 6.5.2 (numbered page 101) of the provisional application.

Applicant believes no fee is due with this response. However, if a fee is due, please charge our Deposit Account No. 18-1945, under Order No. KHEN-P01-001 from which the undersigned is authorized to draw.

Dated: May 25, 2005

Respectfully submitted,

By

Wolfgang E. Stutius
Registration No.: 40,256
ROPES & GRAY LLP
One International Place
Boston, Massachusetts 02110-2624
(617) 951-7000
(617) 951-7050 (Fax)
Attorneys/Agents For Applicant